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(54) Improvements in or Relating to the Control of Mobile Radio Communication Systems

(57) A radio-telephone communications system with a plurality of selectable base stations TR for relaying speech and data radio signals to mobile units is controlled by channel monitors 3 and a master controller 4. The monitors 3 are caused to store digital data describing the quality of audio signals provided by the associated base stations TR in terms of peak and mean values of the amplitude probability density histogram and the master controller 4 interrogates the monitors 3 to select the base stations TR having the best-quality received incoming signal.

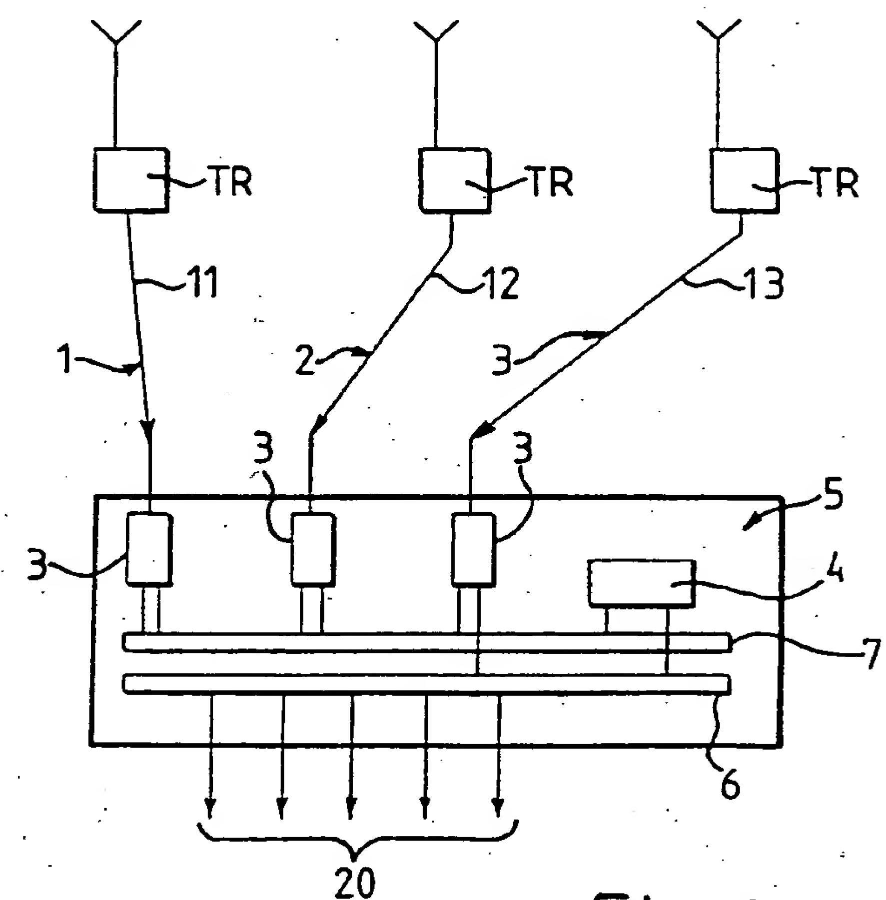

Fig. 1.

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The drawings originally filed were informal and the print here reproduced is taken from a later filed formal copy.

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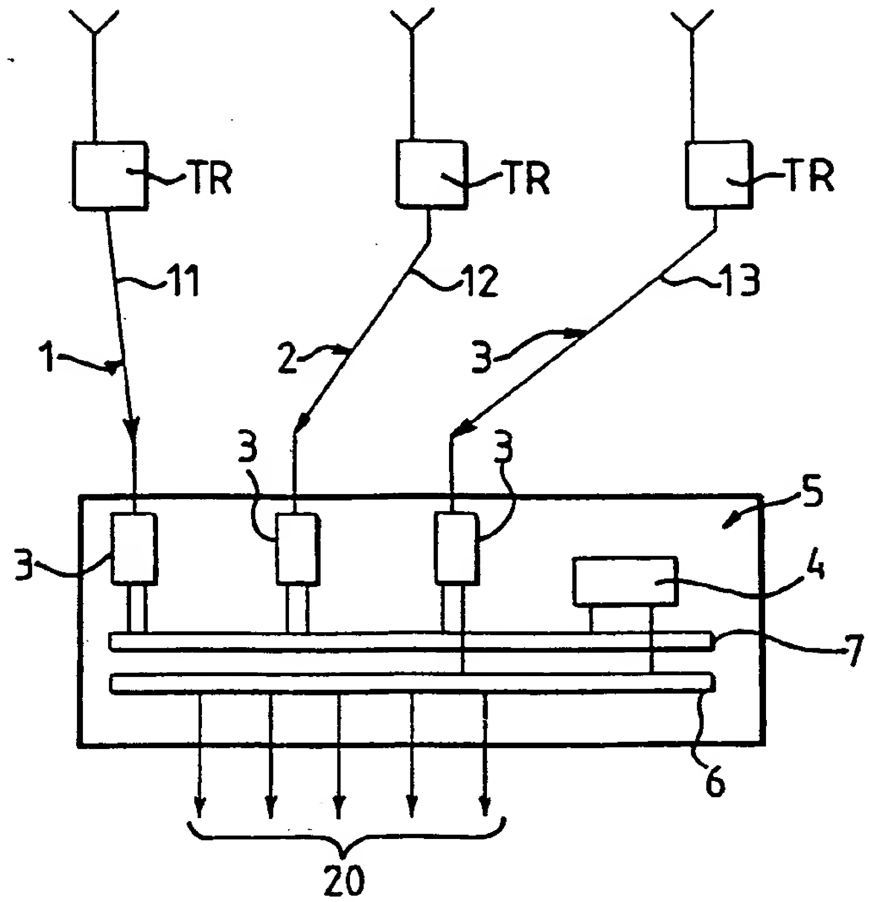
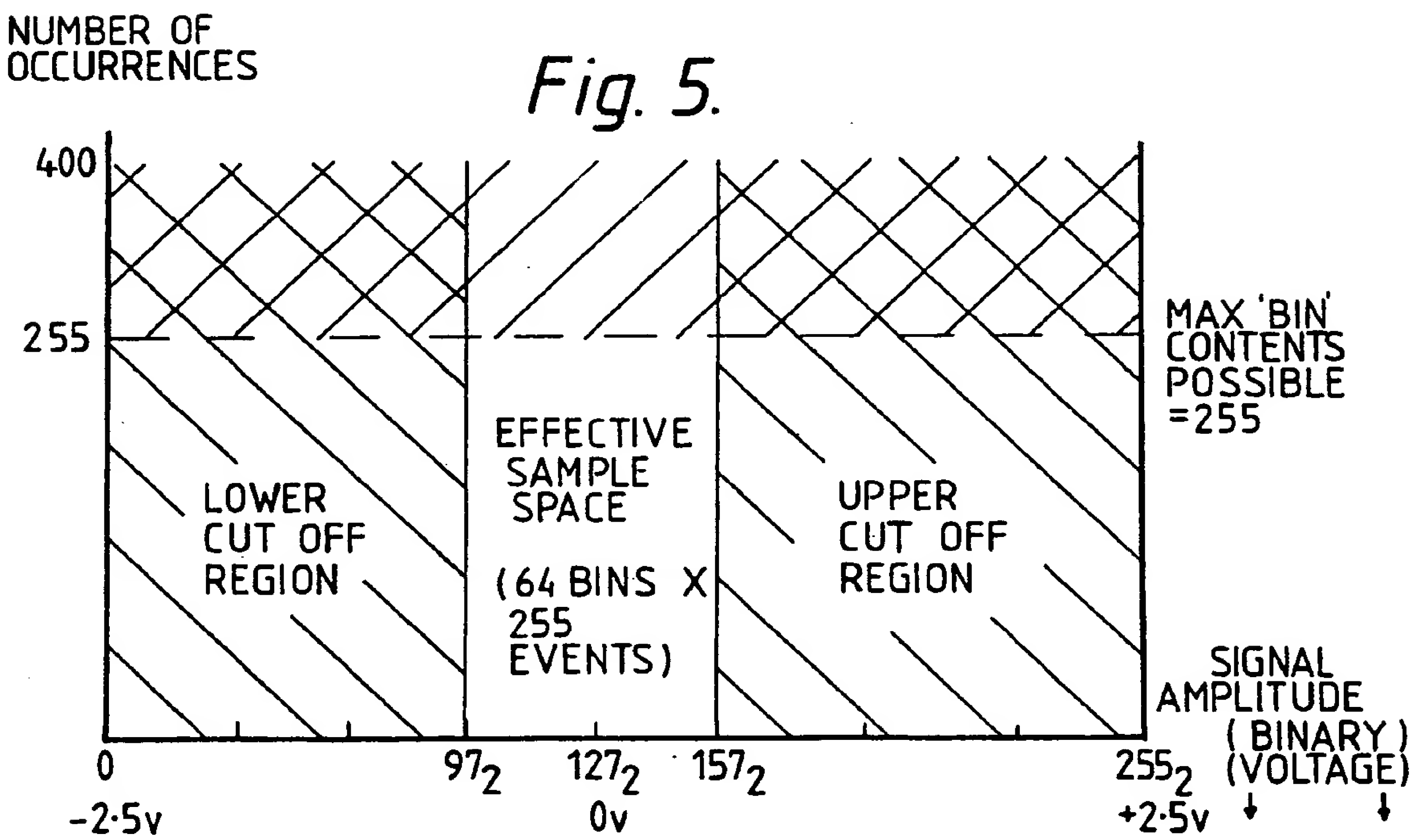
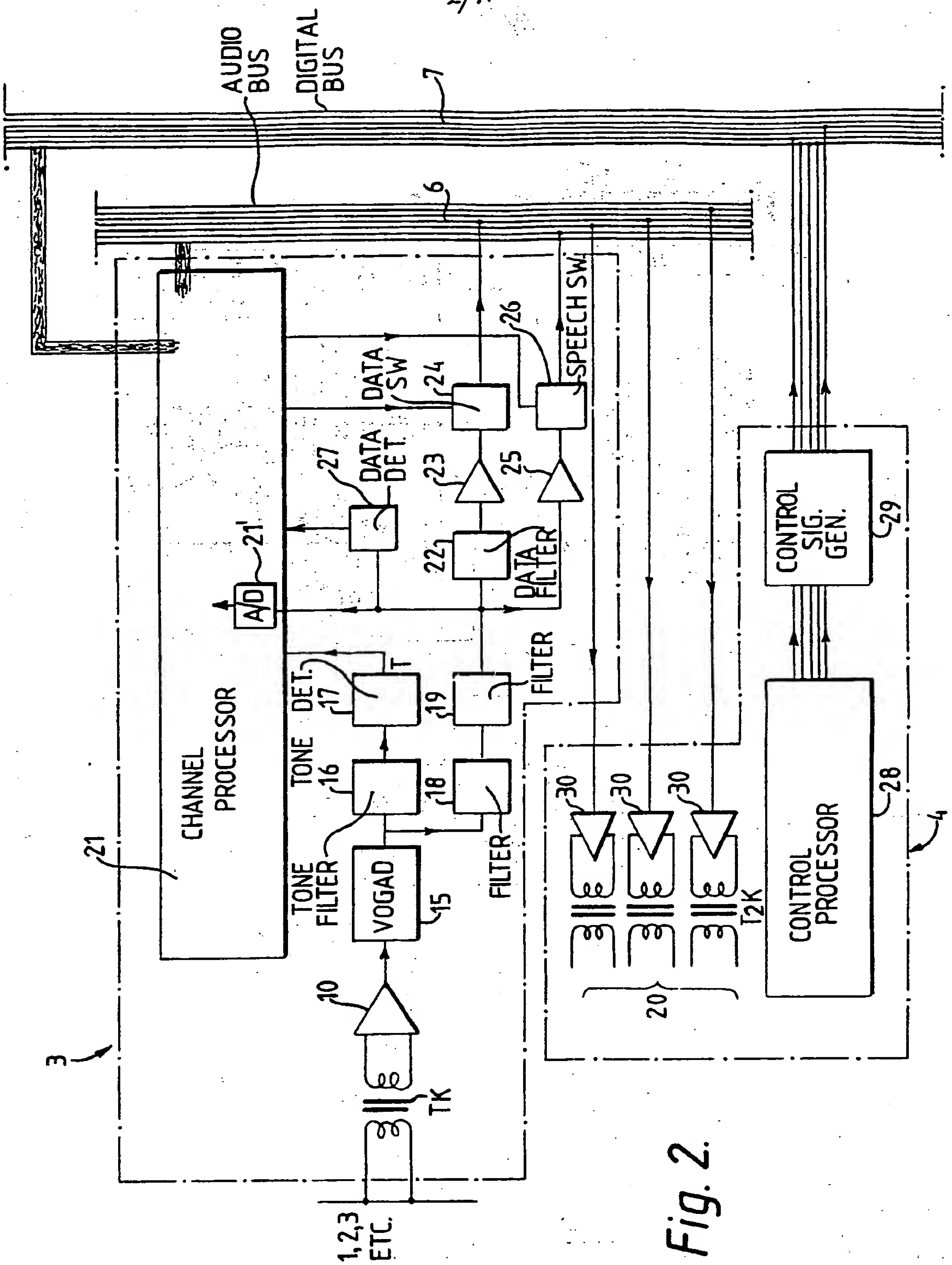


Fig. 1.

Fig. 5.





AUDIO BUS
DIGITAL BUS
7
6
CHANNEL PROCESSOR
21
21'
A/D
DATA DET.
27
DATA SW
24
SPEECH SW.
26
23
25
22
DATA FILTER
TONE DET.
17
T
19
FILTER
TONE
16
18
TONE FILTER
VOGAD
15
FILTER
10
TK
3
1,2,3 ETC.
30
30
30
T2K
20
CONTROL SIG. GEN.
29
CONTROL PROCESSOR
28
4

Fig. 2.

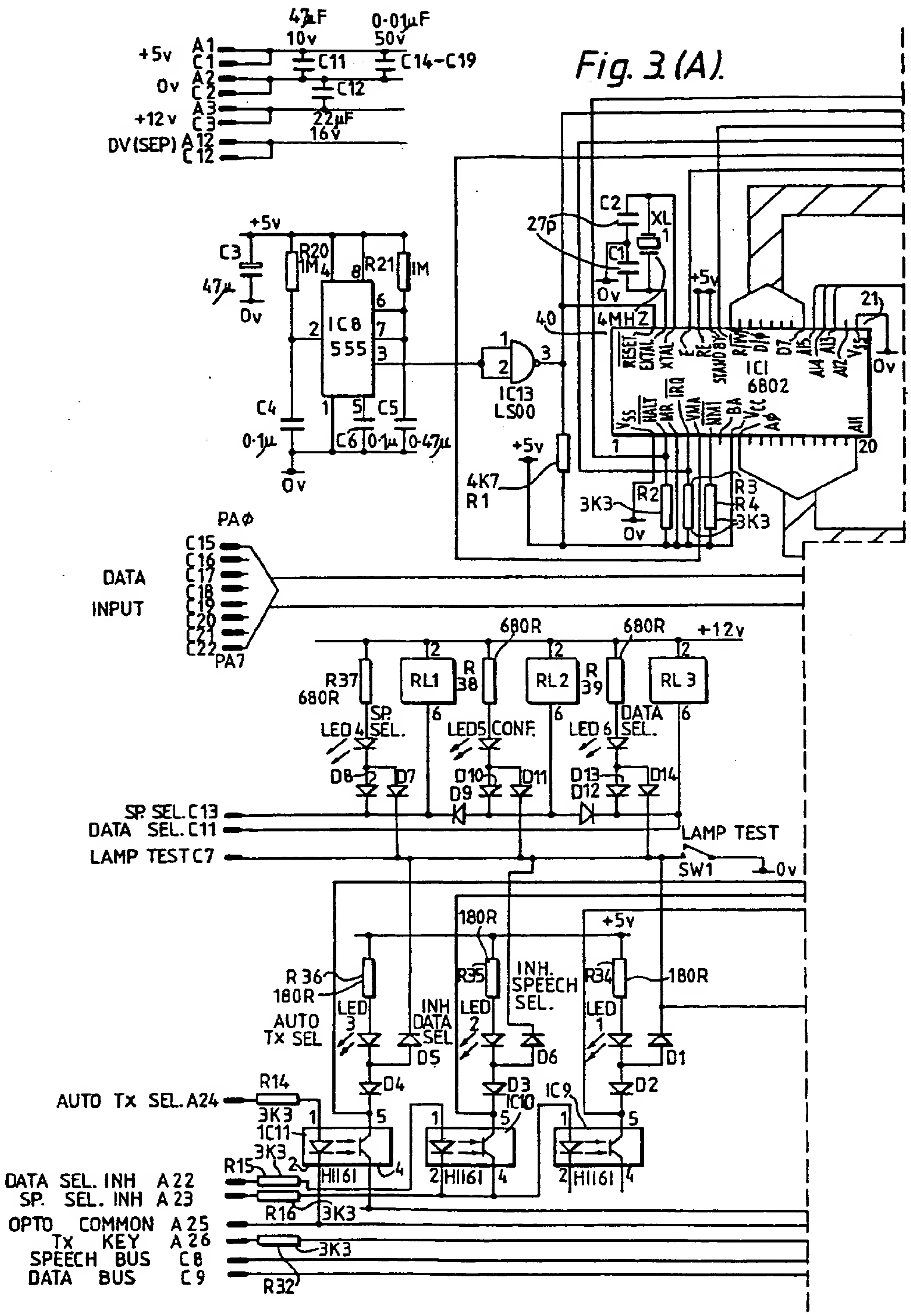


Fig. 3 (A).
+5v
0v
+12v
DV(SEP)
A1
C1
A2
C2
A3
C3
A12
C12
47µF
10v
0·01µF
50v
C11
C14-C19
C12
22µF
16v
+5v
C3
47µ
R20
1M
R21
1M
IC8
555
C4
0·1µ
C6
0·1µ
C5
0·47µ
0v
IC13
LS00
4K7
R1
+5v
C2
27p
XL
1
C1
4MHZ
40
IC1
6802
R2
3K3
R3
R4
3K3
21
20
0v
PAφ
C15
C16
C17
C18
C19
C20
C21
C22
PA7
DATA
INPUT
680R
680R
+12v
R37
680R
RL1
R
38
RL2
R
39
RL3
SP.
SEL.
CONF.
DATA
SEL.
LED 4
LED5
LED 6
D8
D7
D10
D9
D11
D13
D12
D14
SP. SEL. C13
DATA SEL. C11
LAMP TEST C7
LAMP TEST
SW1
0v
180R
+5v
R 36
180R
R35
R34
180R
AUTO
Tx SEL
LED
3
INH
DATA
SEL
D5
LED
2
INH.
SPEECH
SEL.
D6
LED
1
D1
D4
D3
D2
IC10
IC9
AUTO Tx SEL. A24
R14
3K3
1C11
3K3
R15
H1161
H1161
H1161
DATA SEL. INH A22
SP. SEL. INH A23
R16
3K3
OPTO COMMON A25
Tx KEY A26
SPEECH BUS C8
DATA BUS C9
3K3
R32

Fig. 3.(B).

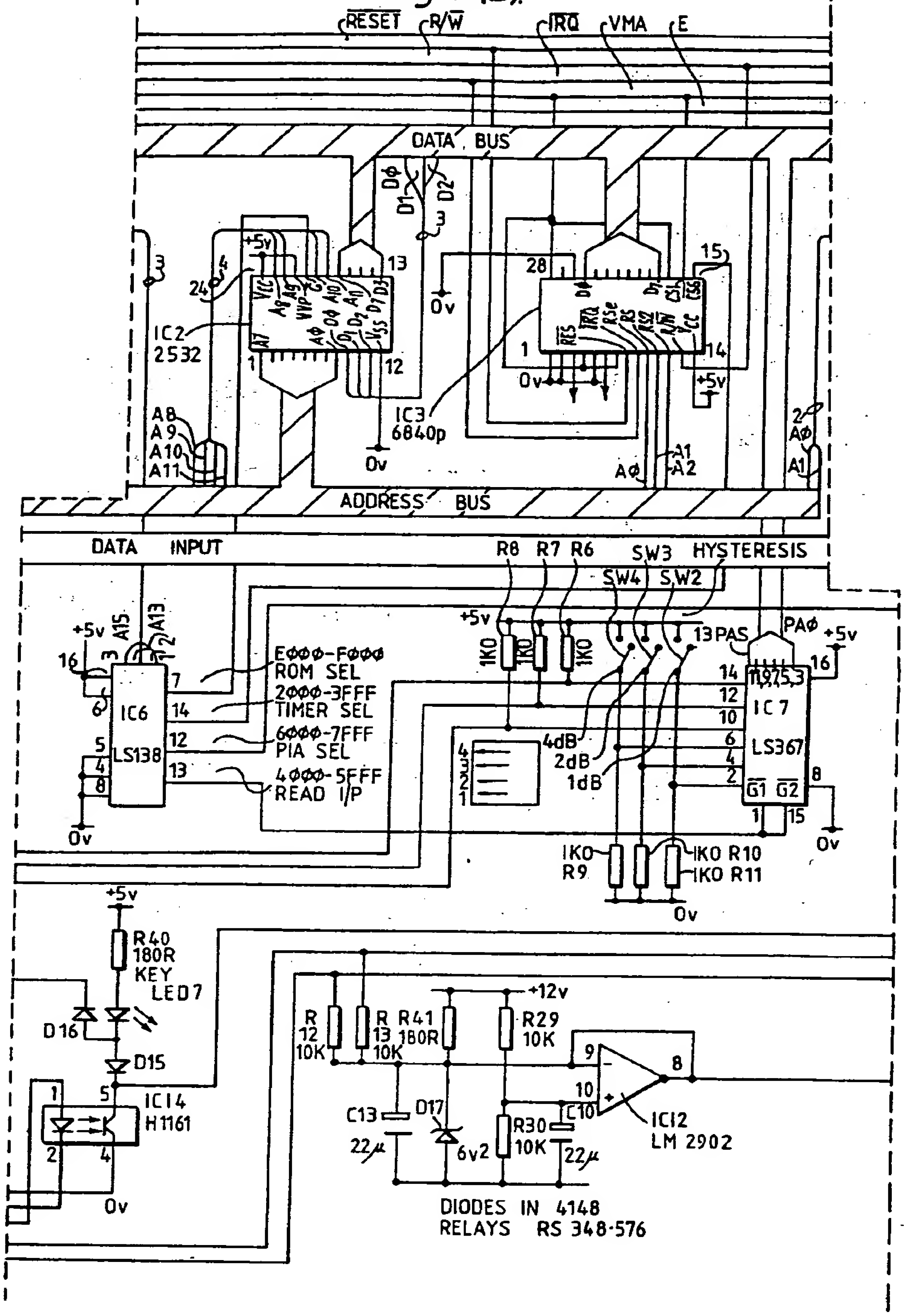
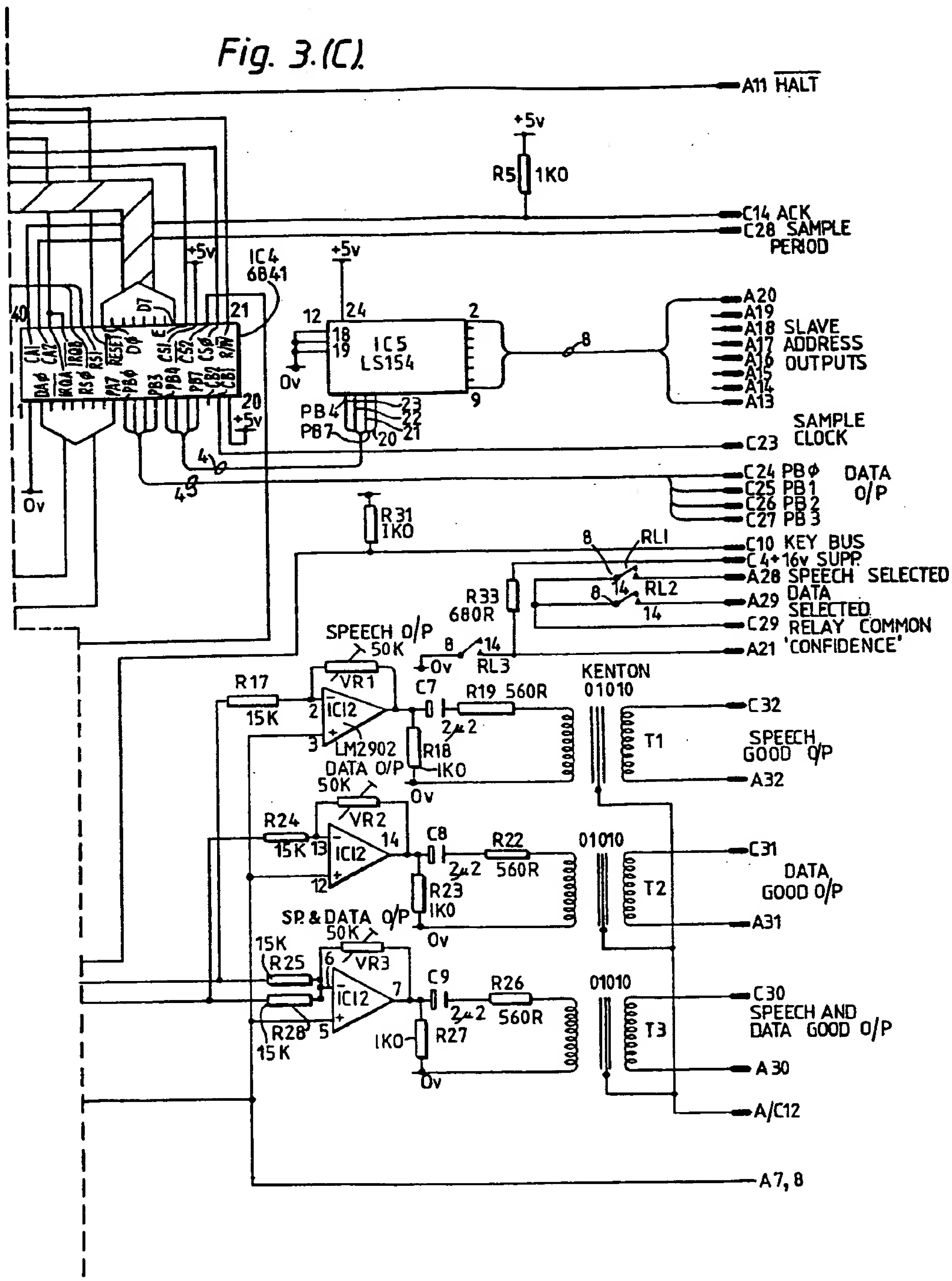

RESET
R/W
IRQ
VMA
E
DATA BUS
IC2
2532
IC3
6840p
ADDRESS BUS
DATA INPUT
IC6
LS138
E000-F000
ROM SEL
2000-3FFF
TIMER SEL
6000-7FFF
PIA SEL
4000-5FFF
READ I/P
R8 R7 R6
SW3 HYSTERESIS
SW4 SW2
13PAS
PA0
IC7
LS367
4dB
2dB
1dB
1K0 R10
1K0 R11
R9
R40
180R
KEY
LED7
D16
D15
IC14
H1161
R12 10K
R13 10K
R41 180R
R29 10K
+12v
C13
D17
6v2
R30 10K
C10
22μ
IC12
LM 2902
DIODES IN 4148
RELAYS RS 348-576

Fig. 3.(C).



A11 HALT
+5v
R5 1K0
C14 ACK
C28 SAMPLE PERIOD
IC4 6841
+5v
IC5 LS154
0v
A20
A19
A18 SLAVE
A17 ADDRESS
A16 OUTPUTS
A15
A14
A13
SAMPLE CLOCK
C23
PB4
PB7
C24 PBØ
C25 PB1
C26 PB2
C27 PB3
DATA O/P
R31 1K0
C10 KEY BUS
C4 +16v SUPP
A28 SPEECH SELECTED
A29 DATA SELECTED
C29 RELAY COMMON
A21 'CONFIDENCE'
RL1
RL2
RL3
R33 680R
SPEECH O/P
50K
VR1
R17 15K
IC12
LM2902
C7
R19 560R
R18 1K0
KENTON 01010
T1
C32
SPEECH GOOD O/P
A32
DATA O/P
50K
VR2
R24 15K
C8
R22 560R
R23 1K0
01010
T2
C31
DATA GOOD O/P
A31
SP. & DATA O/P
50K
VR3
R25 15K
R28 15K
C9
R26 560R
R27 1K0
01010
T3
C30
SPEECH AND DATA GOOD O/P
A30
A/C12
A7, 8
0v

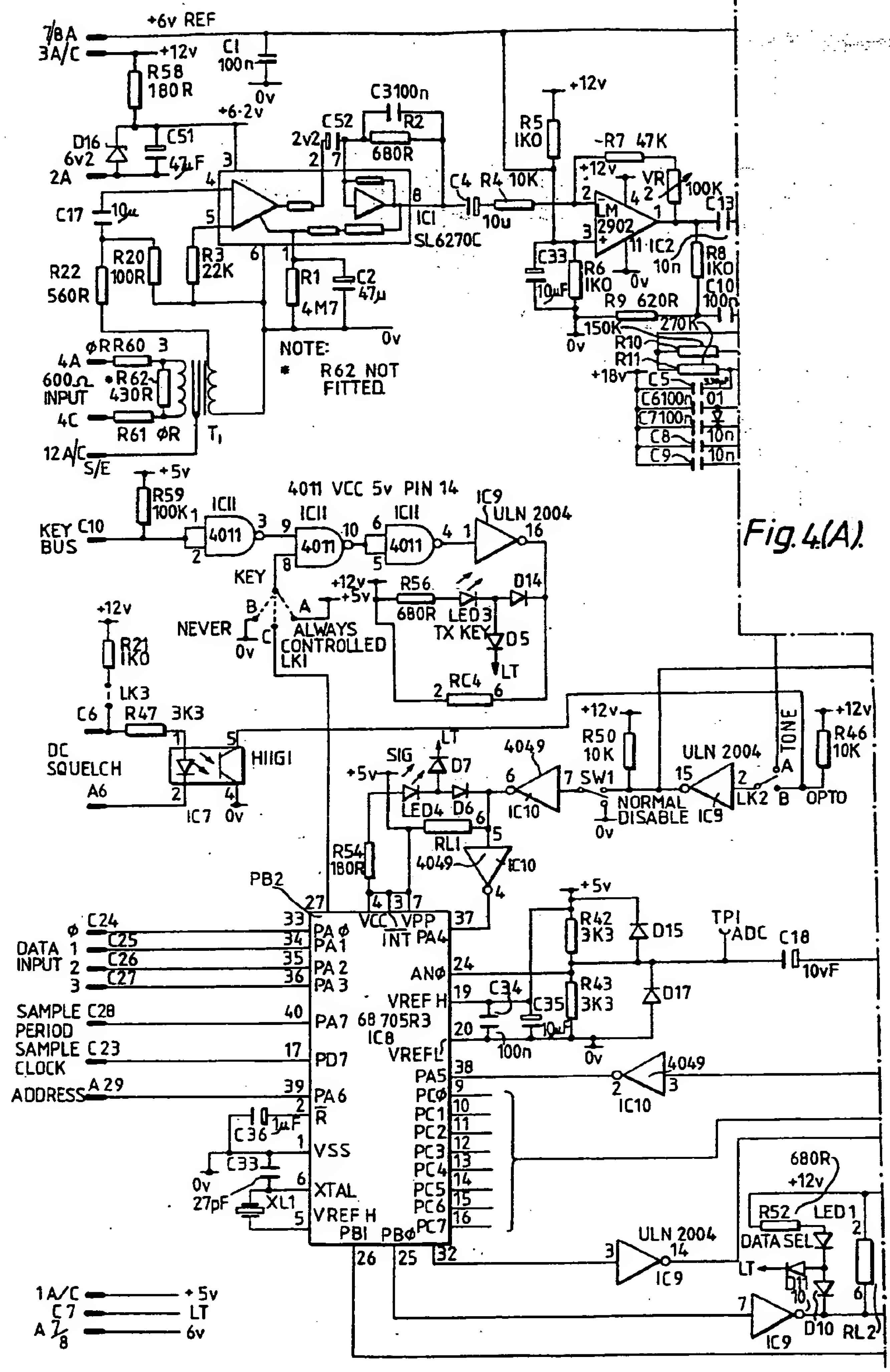

+6v REF
7/8A
3A/C
+12v
C1
100n
0v
R58
180R
+6·2v
C3100n
R2
680R
C52
2v2
D16
6v2
C51
47μF
2A
C17
10μ
IC1
SL6270C
C4
R4 10K
10u
R5
IK0
+12v
R7 47K
VR
100K
LM
2902
IC2
C13
R8
IK0
C10
100n
C33
10μF
R6
IK0
R9 620R
150K
270K
R10
R11
+18v
C5
C6100n
C7100n
C8
C9
10n
R22
560R
R20
100R
R3
22K
R1
4M7
C2
47μ
NOTE:
* R62 NOT FITTED
4A
600Ω INPUT
4C
ØR R60
R62
430R
R61 ØR
T1
12A/C
S/E
+5v
R59
100K
4011 VCC 5v PIN 14
IC11
4011
IC9
ULN 2004
KEY BUS
C10
KEY
NEVER
ALWAYS
CONTROLLED
LK1
R56
680R
LED3
TX KEY
D14
D5
LT
RC4
R21
IK0
LK3
C6
R47 3K3
DC SQUELCH
A6
H11G1
IC7
SIG
D7
4049
LED4
D6
IC10
RL1
SW1
NORMAL
DISABLE
R50
10K
ULN 2004
TONE
LK2
OPTO
R46
10K
R54
180R
PB2
DATA INPUT
C24
C25
C26
C27
SAMPLE PERIOD
C28
SAMPLE CLOCK
C23
ADDRESS
A29
PA Ø
PA1
PA2
PA3
PA7
PD7
PA6
R
VSS
XTAL
VREF H
VCC
VPP
INT
PA4
ANØ
VREFL
PA5
PCØ
PC1
PC2
PC3
PC4
PC5
PC6
PC7
PB1
PBØ
68 705R3
IC8
C36
1μF
C33
27pF
XL1
R42
3K3
D15
R43
3K3
D17
TP1
ADC
C18
10vF
C34
C35
100n
10μ
680R
R52
LED1
DATA SEL
D11
D10
RL2
1A/C
C7
A 7/8
+5v
LT
6v

Fig.4(A).

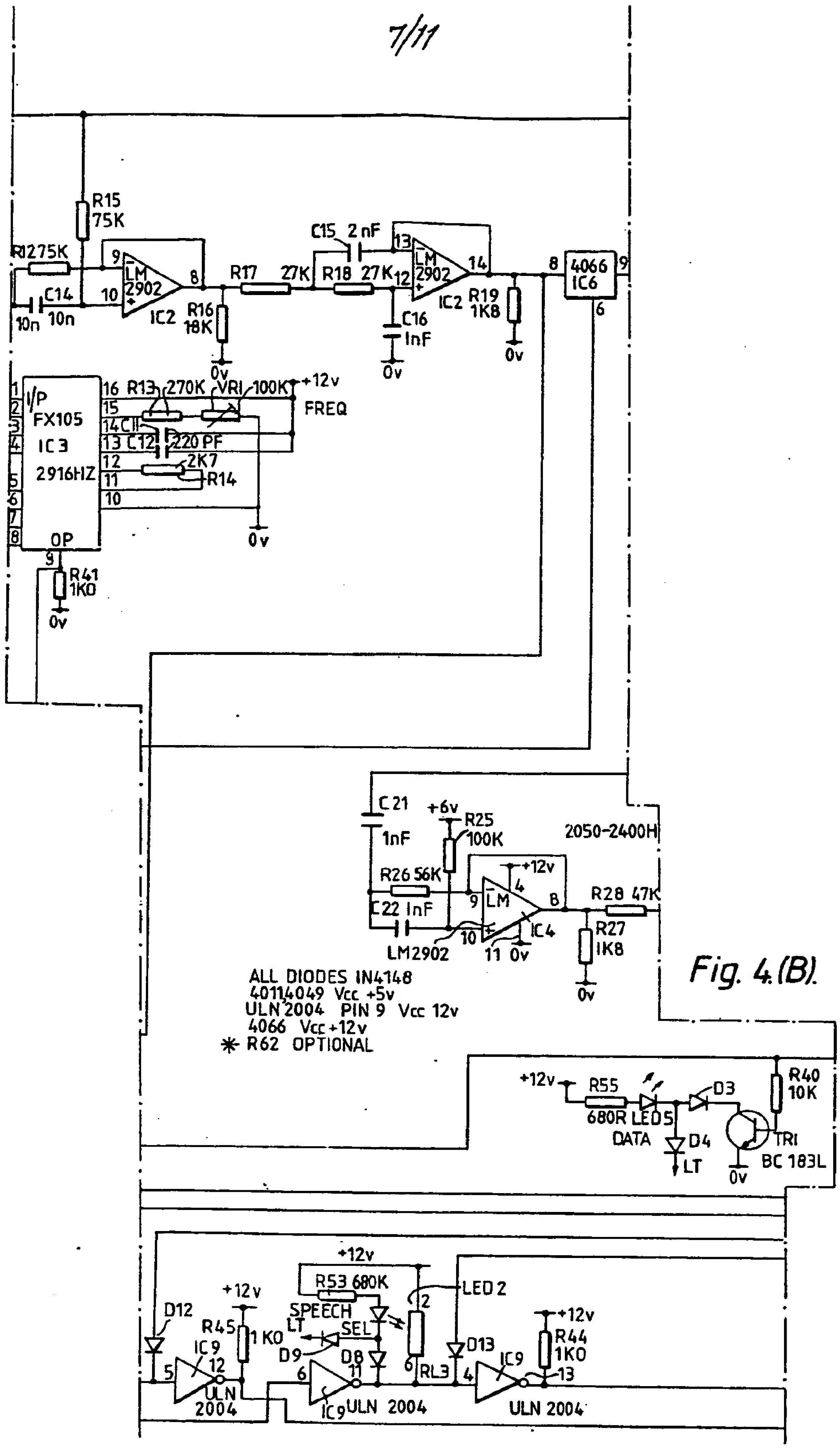

R15
75K
R1275K
C14
10n
10n
LM
2902
IC2
R16
18K
R17 27K
R18 27K
C15 2nF
C16
1nF
R19
1K8
4066
IC6
0v
I/P
FX105
IC3
2916HZ
OP
R13 270K
VR1 100K
+12v
FREQ
C11
C12
220 PF
2K7
R14
R41
1K0
C21
1nF
+6v
R25
100K
2050-2400H
R26 56K
C22 1nF
LM2902
IC4
R28 47K
R27
1K8
ALL DIODES IN4148
40114049 Vcc +5v
ULN 2004 PIN 9 Vcc 12v
4066 Vcc +12v
* R62 OPTIONAL
R55
680R LED5
DATA
D3
D4
LT
R40
10K
TR1
BC 183L
D12
R45
1K0
IC9
ULN 2004
R53 680K
SPEECH
SEL
D9
D8
RL3
LED 2
D13
R44
1K0

Fig. 4.(B).

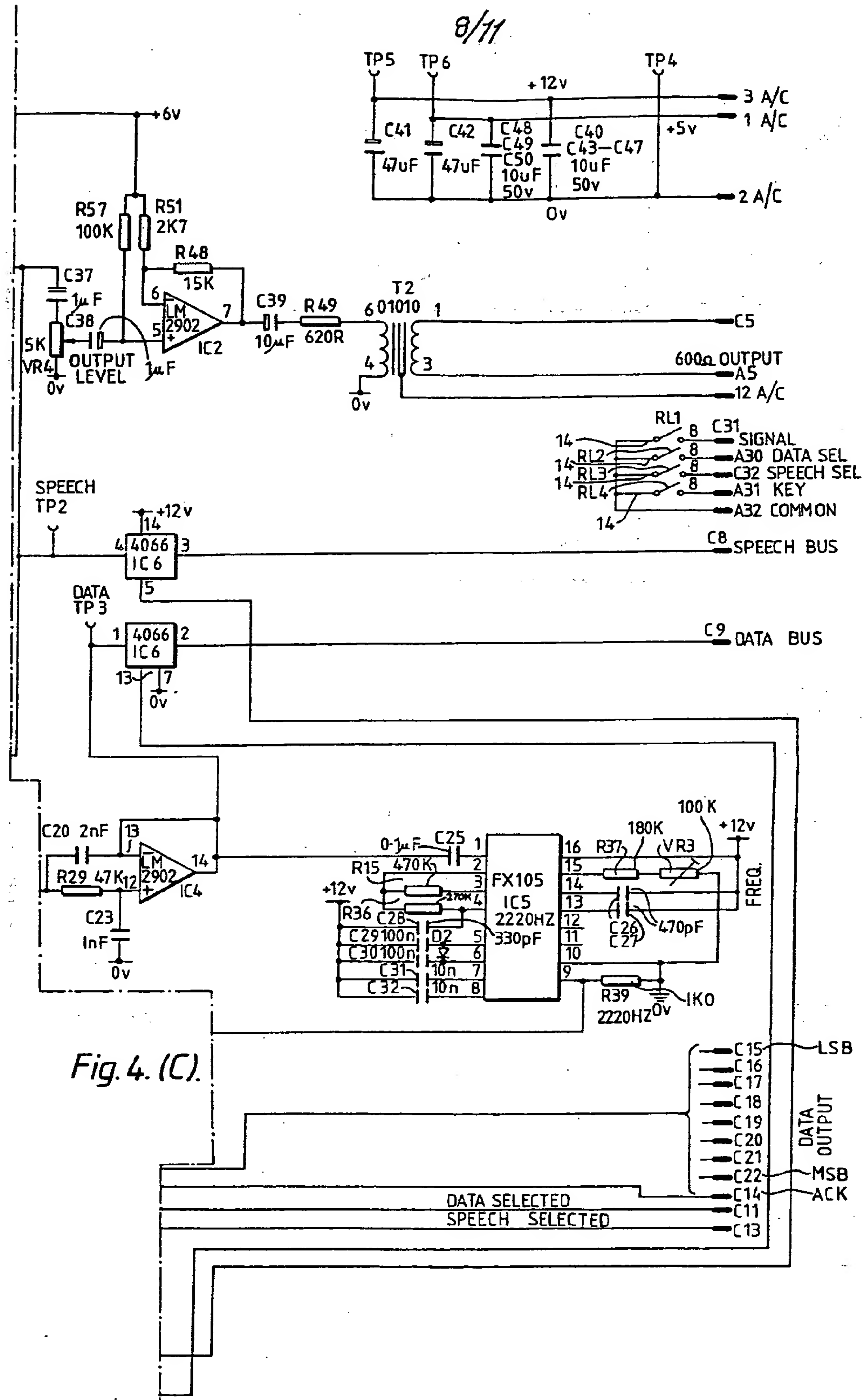

TP5
TP6
TP4
+12v
3 A/C
1 A/C
C41
47uF
C42
47uF
C48
C49
C50
10uF
50v
C40
C43—C47
10uF
50v
+5v
0v
2 A/C
+6v
R57
100K
R51
2K7
R48
15K
C37
1µF
C38
LM
2902
IC2
C39
10µF
R49
620R
T2
01010
5K
VR4
OUTPUT
LEVEL
1µF
0v
C5
600Ω OUTPUT
A5
12 A/C
RL1
C31
SIGNAL
RL2
A30 DATA SEL
RL3
C32 SPEECH SEL
RL4
A31 KEY
A32 COMMON
SPEECH
TP2
4066
IC6
C8
SPEECH BUS
DATA
TP3
C9
DATA BUS
C20 2nF
R29 47K
C23
InF
IC4
0-1µF
C25
R15
470K
+12v
R36
C28
C29 100n
C30 100n
C31
C32
D2
10n
FX105
IC5
2220HZ
330pF
R37
180K
VR3
100K
FREQ.
C26
C27
470pF
R39
2220HZ
IK0
C15
LSB
C16
C17
C18
C19
C20
C21
C22
MSB
C14
ACK
DATA
OUTPUT
DATA SELECTED
SPEECH SELECTED
C11
C13

Fig. 4. (C).

Fig. 6.(A).
PURE NOISE INPUT

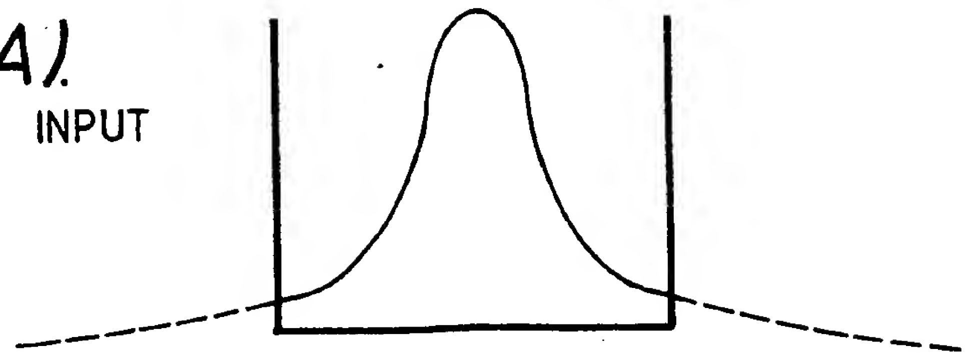


Fig. 6.(B).
PURE TONE INPUT

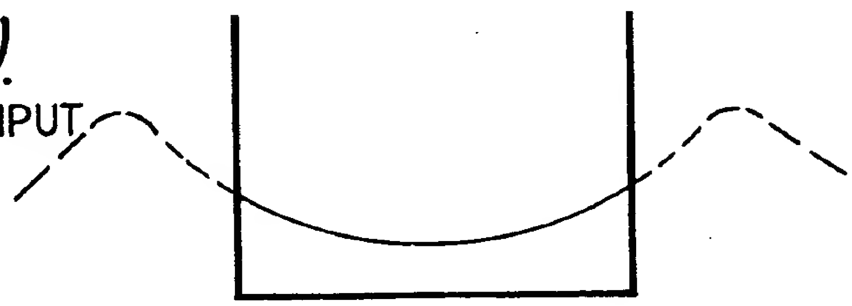


Fig. 6.(C).
TONE AND NOISE

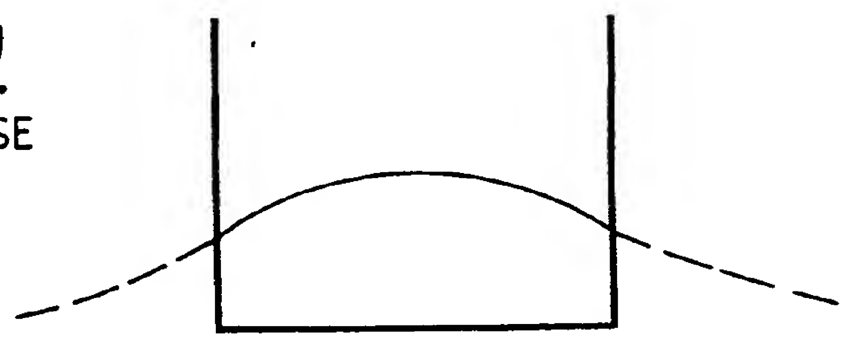
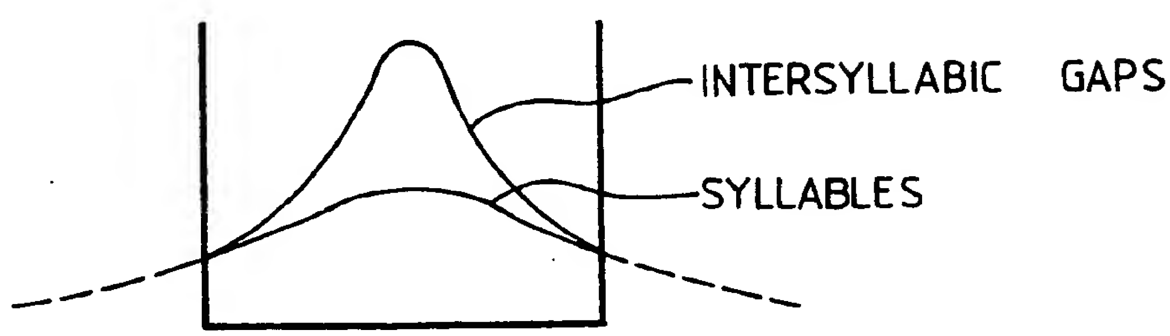


Fig. 6.(D).
SPEECH INPUT



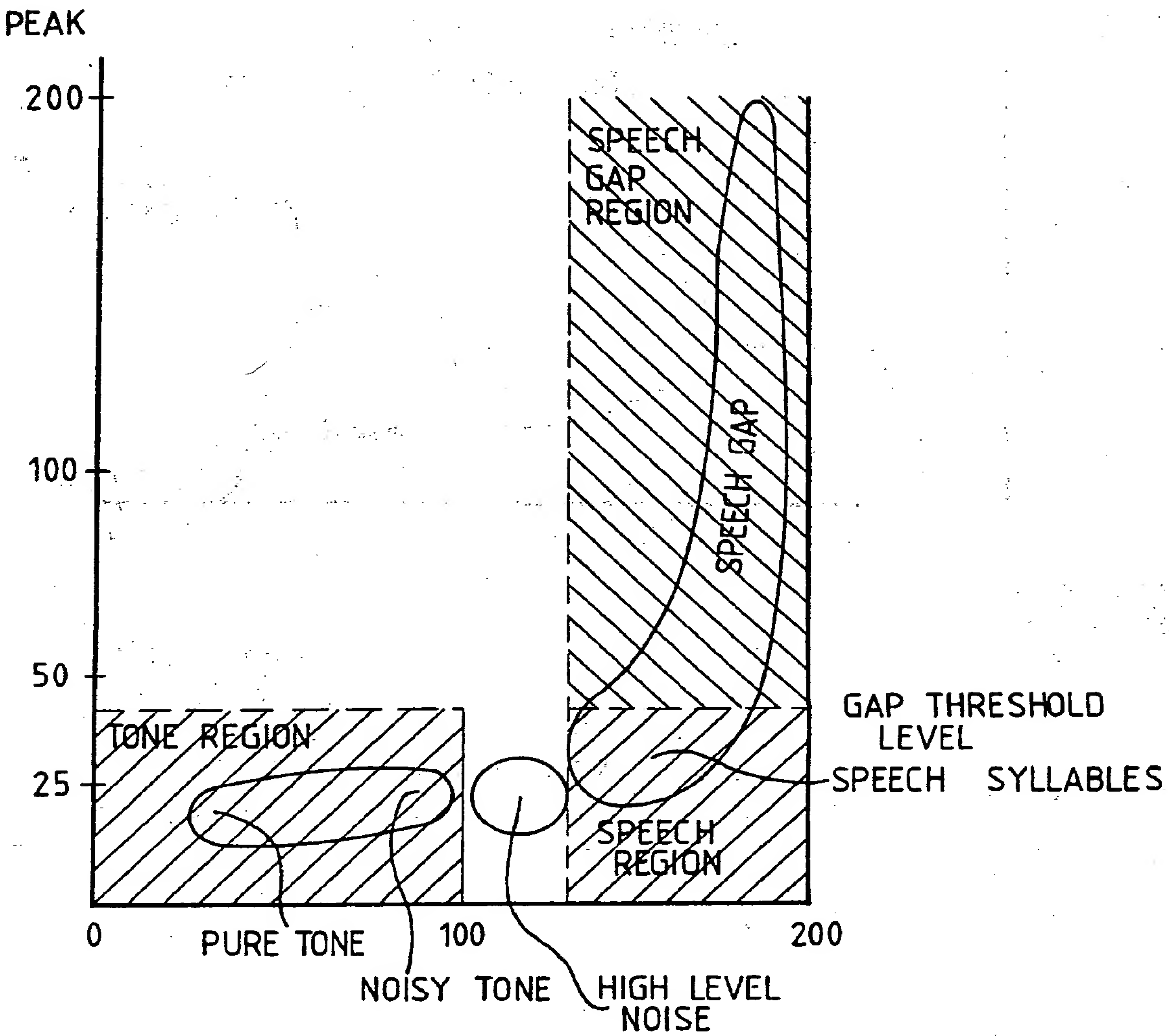


Fig. 7 TYPICAL PEAK/MEAN REGIONS OCCUPIED BY VARIOUS SIGNALS.

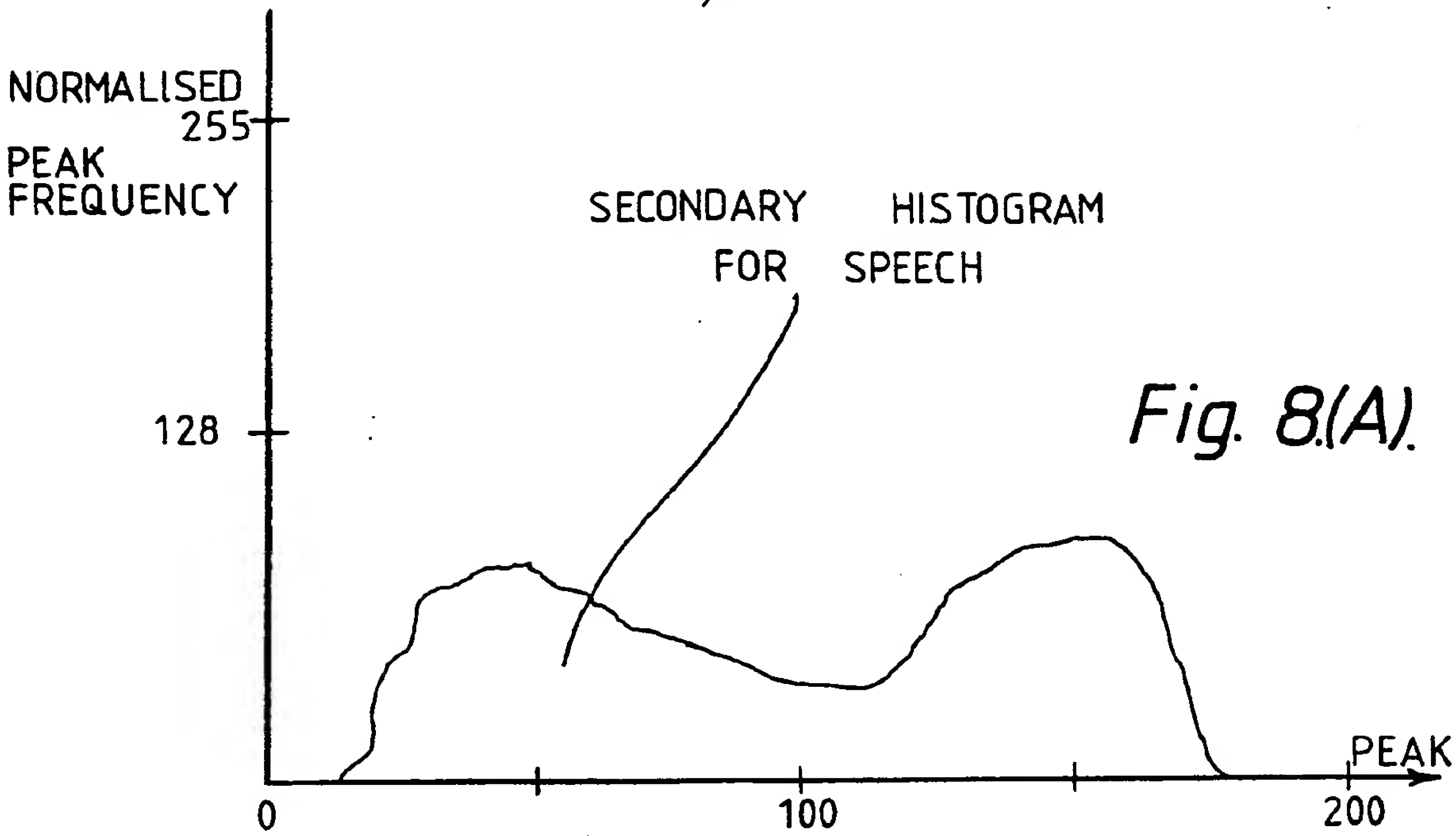
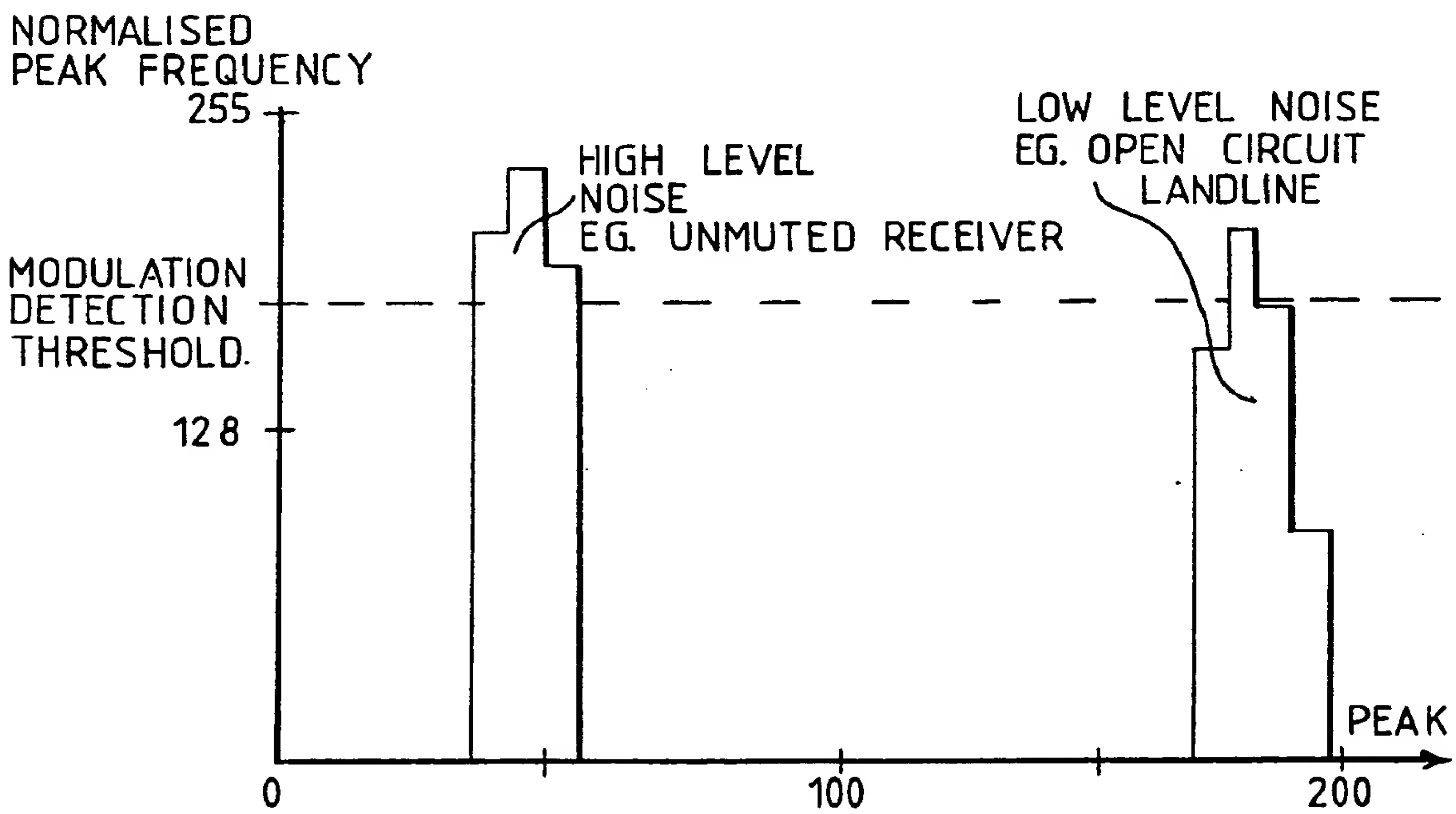


Fig. 8(A).



SECONDARY HISTOGRAM FOR VARIOUS LEVELS OF PURE NOISE.

Fig. 8.(B).

SPECIFICATION

Improvements in or Relating to the Control of Mobile Radio Communication Systems

The present invention relates to mobile radio communications systems and, more particularly, to the control of multiple base stations in radio-telephone systems.

Various problems arise with the allocation of carrier frequencies to different base stations in an attempt to increase the area covered by mobile radio transmission with switchable channels. Conversely, by operating on one common carrier frequency different base stations will exhibit different reception qualities and interference can occur. There is a need for improved control which will automatically select the best-received signal.

In accordance with the present invention, a mobile radio communications system includes a plurality of base stations—receivers or transmitter/receivers—and employs a control method which monitors audio signals from all the base stations simultaneously, provides digital data describing the quality of the audio signals and enables the comparative assessment of such data for the selection of the line or channel with the best quality signal. The digital data describing the signals is derived by a technique which ascertains the amplitude probability density of each signal. The description data is broadly equivalent to the signal to noise ratio of the corresponding reception.

The control method preferably utilises a master controller which controls the operation of channel monitors which receive the individual audio signals, conveniently e.g. via telephone lines or UHF microwave links, from the base stations. The master controller can cause each channel monitor to perform a sequence of sampling operations designed to provide 8-bit or 16-bit digital "words" in a memory describing its associated audio signal in terms of statistical parameters. The master controller can then vote or select the best quality signal and can switch channels to provide optimum results subject to certain constraints.

In contrast to analogue or carrier-based prior art selection systems control in accordance with the present invention provides a number of advantages. Different types of base station and transmissions can be used in the overall system and no modification is necessary to the base station since the control is effective at the audio output stage. Because the control is effected at a central location the geographical location of the base stations does not affect the assessing operation and the degradation of signals over different lengths of telephone line will be also assessed.

Control means in accordance with the invention comprises channel monitors for receiving audio signals from a number of stations, each channel monitor having a channel processor which converts its audio signal into digital signals and performs operations on the digital signals to ascertain the the shape of the amplitude probability density histogram of the audio signal. Preferably the shape is ascertained by stored "words" representing the peak and mid-quartile mean values of the signal's histogram. A master controller provides digital control signals for all the channel monitors and serves to compare the stored words to assess the best-received signal.

The channel monitors and the master controller intercommunicates via a digital and an audio bus. Each channel monitor has further devices to identify the nature of the audio signal i.e. whether speech, noise, data or an optional "receiver muted" signal tone. The channel monitors and master controller preferably employ microprocessors programmed to perform, inter alia, the aforesaid functions. This enables the control means and the communications system to be readily adapted to said particular needs.

The invention may be understood more readily, and various other aspects and features of the invention may become apparent, from consideration of the following description.

An embodiment of the invention will now be described, by way of example only, with reference to the accompanying drawings wherein:

Figure 1 is a block diagram of a typical mobile radio communications systems utilising control means constructed in accordance with the invention.

Figure 2 is block diagram illustrating part of the control means made in accordance with the invention;

Figure 3 is a more detailed representation of the master controller of control means;

Figure 4 is a more detailed representation of one of the channel monitors of the control means; and

Figures 5 to 8 are curves depicting various relationships pertaining to the invention.

As shown in Figure 1 a number of transceivers TR, which may be in fixed locations, receive data and/or speech information (FM or AM) by radio transmissions. The transceivers TR would normally operate at similar frequencies and are notionally associated with respective channels 1, 2 and 3. Three such transceivers TR are shown merely by way of example. The demodulated or detected information is conveyed as audio signals to a central control means 5 which serves to select the best-received signal to relay to one or more users or other equipment via output lines 20. As illustrated, the audio signals are fed to the control means 5 via cables 11, 12, 13 but other methods such as further radio links can replace the cables 11, 12, 13. As is known, the transceivers TR may provide recognisable muting tones when quiescent i.e. when not receiving radio transmissions. The control means 5 comprises individual channel monitors 3 and a master controller 4. The monitors 3 and the controller 4 inter-communicate via a common audio bus 6 and a common digital bus 7.

Figure 2 depicts in schematic form one of the channel monitors 3 in conjunction with the

controller 4. Figure 3 is a more detailed circuit representation of the controller 4 of Figure 2 and Figure 4 is a more detailed circuit representation of one of the monitors 3 of Figure 3. As shown in Figure 2, one monitor 3 is provided for each channel and each monitor comprises a channel processor 21. As shown in Figure 2, the channel audio input, e.g. from the associated cable 11, 12, 13 passes via a 600 ohm balanced transformer TK to an amplifier 10 and a "Vogad" 15. The amplifier 10 and Vogad 15 provide a pre-determined amplitude signal for further processing. The amplifiers 10 and Vogads 15 for all the channels are designed to provide the same predetermined amplitude signal for all the channels. This audio signal is then assessed as to whether it is the mute tone or information. A narrow band pass filter 16 selects the mute tone from the output of the Vogad 15 and feeds a detector 17 which provides a signal e.g. to inhibit selection of the the channel processor 21. The channel processor 21 itself can take the form of an integrated circuit, more particularly, an EPROM microprocomputer device of the type MC68705 R3 (Motorola). Further filters 18, 19 select the expected speech and data information from the full bandwidth of the channels and signals are generated which are indicative of the presence of the data or speech. The audio-information signal, whether speech of data, is fed to an analogue/digital converter 21′ which is part of the channel processor 21 and the digital signals are processed as described hereinafter. The analogue signal fed to the converter 21′ of the processor 21 is also fed to a data channel composed of a data filter 22 and amplifier 23 and a switch 24 and to a speech channel composed of an amplifier 25 and a switch 26. In the presence of data or speech the appropriate switch 24, 26 can be gated by control signals from the processor 21 under control of the controller 4 to allow the data to be fed to the audio bus 6 if the channel in question is the best-selected one.

An additional detector 27 senses data information and provides an alert signal to the processor 21.

The master controller 4 is composed of a control processor 28 and a control signal generator 29 which provide digital control signals to all the channel processors 21 via the digital bus 7. A microcomputer device MC 6802 (Motorola) may serve as the processor 28 and the generator 29. As mentioned previously the data or speech from the best-selected channel is passed to the audio bus 6 and thence passes via amplifiers 30 and 600 ohm balanced transformers T2K to the output lines 20 as speech data or combined outputs.

The processor 28 and generator 29 provide digital signals to the individual channel processors 21 to cause the processors 21 to perform a sequence of operations under the control of programmes. Each processor 21 is caused to ascertain the amplitude probability density of the signal feeding the analogue to digital converter 21′ unless inhibited by the detection of the mute tone.

In essence, each signal is assessed according to a number of pre-determined amplitude levels, the information is stored in digital form in a memory of the processor 21 and values are calculated to represent the shape of the associated Amplitude Probability Density Histogram. The values are taken as an indication of the signal to noise ratio of the associated transceiver TR. This sequence occurs during a sampling mode during which the control processor 28 serves to cause the channel processors 21 to sample their signals and store 8-bit words describing the signal. A typical sample period would be about 50—70 milli seconds wherein some 400 samples may be taken. The control processor 28 then interrogates each channel processor 21 in sequence to select the channel processor 21 with the stored values indicating the best signal to noise ratio. The control processor 28 then performs a control operation typically in 20 milli seconds and in accordance with one or more control algorithms stored as a programme and the best-selected signal can be routed to the lines 20. The analysis of the signals will now be described in more detail. The audio signals fed to the individual channel monitors 3 may have widely differing mean voltage levels and the "Vogads" 15 normalise the signals to a known amplitude level so that the noise content of the signal can be taken as its relative signals to noise ratio. The normalised signals are then analysed by the respective processors 21. Each signal is sampled at a sampling rate of 8 KHz by the converter 21′ and the samples are represented as 8 bit words with a resolution of 19.5 mV per increment. As mentioned, 400 samples are taken and stored every sampling period and these samples are evaluated by a process known as "binning". Basically for each 19.5 mV increment of input signal possible within the range —0.62 to +0.62 volts, an 8 bit wide memory location (or "bin") is reserved, giving 64 bins in total. For each occurrence of a voltage corresponding to a particular bin, the contents of that bin are incremented by one. Voltages outside the range —0.62v to +0.62v do not cause an increment and hence their occurrence is ignored. After the 400 samples have been taken the process is complete and the result is a stored histogram of Input Voltage against number of occurrences of that voltage. Since it is unlikely that any one bin will contain the whole 400 samples (i.e. a D.C. level or a signal of amplitude less than 19.5 mV, including Noise) it is possible to limit the contents of each bin to 255 (a convenient 8 bit definable number). In the event of a bin reading 255 the next bin (having an address higher by one) is incremented to assure that the total number of increments received by the 64 bins is preserved. A sample space graphically depicted in Figure 5 is thus established.

A general statistical theorem, known as the

Central Limited theorem, states that the probability distribution of any quantity which arises as the sum of the effects of a large number of separate contributions is the Gaussian or Normal distribution. Electrical Noise is one such quantity and the processors 21 produce such a distribution when presented with electrical Noise. It can be expressed by the function:

$$F(x) = \left(\frac{1}{\sigma\sqrt{2\pi}}\right) \exp\left[\frac{-x^2}{2\sigma^2}\right]$$

where
σ=Standard Deviation
and
$\sigma^2=\bar{x}^2$=Root means square value of the original waveform
So
σ^2=Vnoise (Rms) volts.

For a standardised Normal Variable the maximum value of F(x) can be described by:

$$F(x)max = \frac{1}{\sigma\sqrt{2\pi}}$$

Therefore for any normal distribution
Peak$\propto 1/\sigma$
or

$$Peak \propto \frac{1}{\sqrt{V\ noise}}$$

Hence the "Peak" as calculated can be used as a measure of the noise content of a waveform.

Figure 6 depicts various histogram waveforms.

From the stored histograms in the processors 21, two values known as the "Peak" and "Quartile mean" can be derived and stored for each stored histogram in order to obtain comparative measure of the noise content of the corresponding input signal. The peak value is calculated using a "moving window" technique designed to smooth the curve of the histogram, thus reducing the likelihood of a spurious result from influencing the selection procedure. A 4-bin wide "window" is moved across the total 64 bins in increments of one bin. For each increment the sum of the 4 bins is then taken as the "peak" value. To evaluate the quartile mean, the contents of all 64 bins is divided by 2. This gives a measure of the total area beneath the curve between the voltage limits.

The master control processor 28 reads and stores the values for peak and quartile means for each channel processor 21. If the value falls in a certain region of Peak/Quartile mean relationship the processor 28 assumes the signal is of a type associated with that region.

Figure 7 illustrated how different signal types, with varying quantities of noise have unique ranges of values for the peak and quartile mean associated with them. It is this feature which enables the type and comparative noise content of a signal to be determined. The "undefined" areas are included to ensure that a signal which inhabits a shared region of Peak/Quartile means relationship is not allowable to feature in the signal type decision process. This process is a democratic one i.e. for each channel it is decided whether the signal is speech tone, noise only or an undefined signal and the type is taken to be the one represented by the most channels. Once the signal type has been defined the control programme can select which channel is presenting the lowest noise by comparing peak values in the case of speech or quartile means values in the case of tone. Syllables themselves induce a low peak. However, when the syllable has finished the "gap" induces a high peak (with a fast attack) which then slowly decays due to the action of the Vogad 15. The arrival of the next syllable rapidly forces the peak back down to a low level. Comparison of clean and noisy Histogram shows that the "word" peaks are not significantly different, whereas the value of gap peaks are much greater for clean than for noisy speech. The control processor 28 software analyses the gap peaks of all the channels and selects the one which persistently portrays the highest value and consequently the lowest noise. Thus, with speech the selection is effectively made by comparing the residual channel noise occuring during inter-word or inter-syllabic gaps. The twin peaks of the histogram of a tone (see Figure 6b) is a characteristic of the probability analysis of a waveform of a repetitive nature. However, the Vogad 15 is adjusted such that the peaks occur outside the voltage cut-off points, leaving only the centre "trough" for analysis. This has the effect that for a good quality signal the majority of occurrences arise with the two peaks and consequently less occurrences are found in the analysed region, thus reducing the quartile mean.

The control means can provide a variety of visual and/or signal indications to a user which are not shown in the drawings and the control processor 28 and/or the channel processors 21 can be programmed to cope with a variety of user demands of which the following are just typical examples.

1. Bias can be provided as between channels so that for equal equivalent signal to noise ratio on two channels one is given preference.

2. Hysterisis can be provided as between channels so that channel change over will not occur unless the difference between the equivalent signal to the noise ratios is greater than some pre-determined threshold.This effect can be accomplished by artificially enhancing the parameters describing the currently selected channel. A certain percentage of the original peak and quartile the mean values associated with the selected channel are either added in the case of peak or subtracted in the case of mean, to make the channel currently selected hold an advantage in the "best value" comparison procedure. The percentage can be varied over the range 0.6dB.

3. Data information may be locked so that

channel switching will not occur if certain data is present.

4. Emergency or coded data information may be given preference at all times.

5. Confidential channels can be established and maintained as such.

6. Where a selected base station is to used also as a transmitter in the case of "talk-through" automatic keying can be provided.

7. A facility may be provided whereby a mute tone signal is not required, because the system is able to detect the absence of speech or tone modulation. If no modulation is detected, on a given channel then no further selection is made on that channel. This may be accomplished by using the peak values of the Amplitude probability Histogram, to form a second histogram, in which the relative frequency of occurrence of a given peak determines the height of a given ordinate. Pure noise signals may have low peaks (i.e. a muted receiver) or high peaks (i.e. line noise) but successive peaks of a given noise level tend to be closely located around an average value. Only modulated signals give large variations in successive peak values. Thus the unmodulated signal may be identified by setting a threshold level for peak frequency as represented in Figure 8.

CLAIMS

1. A method of controlling a mobile radio communication system by simultaneously monitoring signals provided by base stations, describing the quality of the signals by digital data and assessing the data to select the base station providing the best quality signal.

2. A method according to claim 1, wherein the digital data is derived by assessment of parameters associated with the amplitude probability density of the signals.

3. A method according to claim 1, wherein the signals are sampled repetetively to provide 8-bit or 16-bit digital words describing the signals in terms of statistical parameters.

4. A method according to claim 1, wherein the digital data describing the signals represents the shape of the amplitude probability density histogram of each of the signals.

5. A method according to claim 4, wherein the digital data describing the signals represents the peak and mid quartile mean values.

6. A method according to any one of claims 1 to 5, wherein the monitored signals are audio signals.

7. A method according to any one of claims 1 to 5 and further comprising assessing whether the signals provided by the base stations represent speech, tone, noise or data information.

8. A method of controlling a mobile radio communications system substantially as described with reference to any one or more of the Figures of the accompanying drawings.

9. A control means for a mobile radio communication system comprising base stations in the form of receivers or transmitters/receivers; said control means comprising channel monitors for receiving signals from the stations and for converting the signals into digital data describing the quality of the signals, and a master controller for interrogating the channel monitors to assess the best-received signal and enable the selection of the appropriate base station.

10. Control means according to claim 9, wherein the channel monitors each employs a channel processor which stores digital words representing the peak and mid-quartile values of an amplitude probability histogram of the associated signal under control of the master controller.

11. Control means according to claim 9 or 10, wherein the channel monitors employ devices serving to normalise the amplitudes of all the signals for their conversion.

12. Control means according to any one of claims 9 to 11, wherein the channel monitors receive audio signals from the base stations, the channel monitors and the master controller inter-communicate via a common digital bus and a common audio bus and the master controller connects the audio bus to outputs to interconnect the selected base station to said outputs.

13. Control means according to claim 12, wherein each channel monitor has devices to identify the nature of the signals provided by the base stations and to gate the signals to the audio bus.

14. Control means according to any one of claims 9 to 13, wherein the digital data is derived as described herein with reference to Figure 5.

15. Control means according to any one of claims 9 to 14, wherein the channel monitors and/or the master controller utilise replaceable pre-programmed or programmable integrated circuit devices.

16. A radio communications system or control means therefot substantially as described with reference to, and as illustrated in any one or more of Figures 1 to 4 of the accompanying drawings.

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